In the Specification:

Please replace the paragraph beginning at page 11, line 22, with the following paragraph:

In one aspect of the present invention, the signal gain may be calculated based on $\gamma_{db} = \mu_g\ (\sigma''_q - \sigma_{th}) + \gamma_n$, such that μ_g is adjusted according to the voicing parameter(s). In other aspects, the voicing parameter(s) may be used to adjust other parameters in the above-shown equation, such as σ_{th} or γ_n, or elements of any other equation used for noise ~~suppresion~~ suppression purposes.

A1

Please replace the paragraph beginning at page 17, line 12, with the following paragraph:

Bandwidth expansion of 60Hz and a white noise correction factor of 1.0001, i.e. adding a noise floor of –40dB, are applied by weighting the auto-correlation coefficients according to $r_w(k) = w(k) \cdot r(k)$, where the ~~weigthing~~ weighting function is given by

$$w(k) = \begin{cases} 1.0001 & k = 0 \\ \exp\left[-\frac{1}{2}\left(\frac{2\pi \cdot 60 \cdot k}{8000}\right)\right] & k = 1,2,\ldots,10 \end{cases}.$$

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Please replace the paragraph beginning at page 17, line 16, with the following paragraph:

Based on the weighted auto-correlation coefficients, the short-term LP filter coefficients, i.e. $A(z) = 1 - \sum_{i=1}^{10} a_i \cdot z^{-i}$, are estimated using the Leroux-Gueguen algorithm, and the line spectrum frequency ("LSF") parameters are derived from the polynomial $A(z)$. The three sets of LSFs are denoted $\mathrm{lsf}_j(k)$, $k = 1,2\ldots,10$, where $\mathrm{lsf}_2(k)$, $\mathrm{lsf}_3(k)$, and $\mathrm{lsf}_4(k)$ are the LSFs for the middle third, last third and look-ahead of each frame, respectively.

A3

Please replace the paragraph beginning at page 20, line 14, with the following paragraph:

Referring to FIG. 2, the encoder 200 further classifies the pre-processed speech signal 207. The ~~calssification~~ classification module 230 is used to emphasize the perceptually important features during encoding. According to one embodiment, the three main frame-based classifications are detection of unvoiced noise-like speech, a six-grade signal characteristic classification, and a six-grade classification to control the pitch pre-processing. The detection of unvoiced noise-like speech is primarily used for generating a pitch pre-processing. In one embodiment, the classification module 230

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classifies each frame into one of six classes according to the dominating feature of that frame. The classes are: (1) Silence/Background Noise, (2) Noise-Like Unvoiced Speech, (3) Unvoiced, (4) Onset, (5) Non-Stationary Voiced and (6) Stationary Voiced. In some embodiments, the classification module 230 does not initially distinguish between non-stationary and stationary voiced of classes 5 and 6, and instead, this distinction is performed during the pitch pre-processing, where additional information is available to the encoder 200. As shown, the input parameters to the classification module 230 are the pre-processed speech signal 207, a pitch lag 231, a correlation 233 of the second half of each frame and the VAD information 225.

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Please replace the paragraph beginning at page 22, line 8, with the following paragraph:

Turning back to the speech pre-processor block 210, as discussed above, the noise suppression module 206 receives various voicing parameters from the speech processor block 250 in order to improve the ~~calcualtion~~ calculation of the channel gain. The voicing ~~parametres~~ parameters may be derived from various modules within the speech processor block 250, such as a the classification module 230, the pitch estimation module 232, etc. The noise suppression module 206 uses the voicing ~~parametrs~~ parameters to adjust the channel gains {γch(i)}.

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Please replace the paragraph beginning at page 22, line 14, with the following paragraph:

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As explained above, the goal of noise suppression, for a given channel, is to adjust the gain γ_{ch} such that it is higher or closer to 1.0 to preserve the speech quality for strong voiced areas and, on the other hand, ~~lowring~~ lowering the gain γ_{ch} to be closer to zero for suppressing the noise in noisy areas of speech. ~~Theoratically~~ Theoretically, for a pure voice signal, the gain γ_{ch} should be set to "1.0", so the signal remains ~~unmodifed, on~~ On the other hand, for a pure noise signal, the gain γ_{ch} should be set to "0", so the noise signal is suppressed. In between these two ~~theoratical extrems~~ theoretical extremes, there lies a spectrum of possible gains γ_{ch}, where for voice signals, it is desirable to have a gain γ_{ch} closer to "1.0" to preserve the speech quality as much as possible. Now, since the speech processor block 250 contributes to cleaning or suppressing some of the noise in the voiced areas, the conventional noise suppression process may be relaxed (as discussed below.) For example, referring to FIG. 3, speech sections 302, 304 and 306 that are located between the harmonics in the voiced area have a very low signal-to-noise ratio and as a result the speech sections 302, 304 and 306 are noisy sections of the voiced area. But, it should be noted that the speech processor block 250 contributes to cleaning the noisy speech areas 302, 304 and 306 by applying pitch enhancement. Accordingly, modification of the speech signal by reducing the gain γ_{ch} in such areas may be avoided.

Please replace the paragraph beginning at page 23, line 10, with the following paragraph:

The present invention overcomes the drawbacks of the conventional approaches and improves the gain computation by using other dynamic or voicing parameters, in addition to the SNR parameter used in conventional approaches to noise suppression. In one embodiment of the present invention, the voicing ~~paramters~~ parameters are fed back from the speech ~~precossor~~ processor block 250 into the noise suppression module 206. These voicing ~~parametrs~~ parameters belong to previously processed speech frame(s). The advantage of such embodiment is achieving a less complex system, since such embodiment reuses the information gathered by the speech processor block 250. In other embodiments, however, the voicing ~~parametrs~~ parameters may be calculated within the noise suppression module 206. In such embodiments, the voicing parameters may belong to the particular speech frame being processed as well as those of the preceding speech frames.

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Please replace the paragraph beginning at page 24, line 8, with the following paragraph:

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Yet, in other embodiments, the voicing parameters may be used to modify any of the other parameters in the $\gamma db(i)$ equation, such as γn or σth. Nevertheless, the voicing

parameters are used to adjust the gain for each channel through the calculation of the value of "x" by the noise suppression module 206. For example, in one embodiment, the noise suppression module 206 may use the classification parameters from the classification module 230 to calculate the adjustment value "x". As explained above, in one embodiment, the classification module 230 classifies each speech frame into one of the six classes in accordance to the dominating features of each frame. With reference to FIG. 4, if the frames is classified to be in the unvoiced area 410, $\mu g(i)$ will be 0.39. However, if the frames ~~is~~ are classified as being in the voiced area 420, $\mu g(i)$ will 0.39 + x, and "x" may be adjusted based on the strength of the voice signal. For example, if the voice signal is classified as stationary voiced, the value of "x" will be higher, but for non-stationary voiced classification, the value of "x" will be less.

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